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EXAMINER	
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ART UNIT	**PAPER NUMBER**
2685	14

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Please find below and/or attached an Office communication concerning this application or proceeding.

Office Action Summary	Application No.	Applicant(s)	
	09/977,337	CHOONG ET AL.	
	Examiner	Art Unit	
	Duy K Le	2685	

-- The MAILING DATE of this communication appears on the cover sheet with the correspondence address --

Period for Reply

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE 3 MONTH(S) FROM THE MAILING DATE OF THIS COMMUNICATION.

- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed after SIX (6) MONTHS from the mailing date of this communication.
- If the period for reply specified above is less than thirty (30) days, a reply within the statutory minimum of thirty (30) days will be considered timely.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133). Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any earned patent term adjustment. See 37 CFR 1.704(b).

Status

1)☐ Responsive to communication(s) filed on _____.

2a)☐ This action is **FINAL**. 2b)☒ This action is non-final.

3)☐ Since this application is in condition for allowance except for formal matters, prosecution as to the merits is closed in accordance with the practice under *Ex parte Quayle*, 1935 C.D. 11, 453 O.G. 213.

Disposition of Claims

4)☒ Claim(s) 1-17 is/are pending in the application.

4a) Of the above claim(s) _____ is/are withdrawn from consideration.

5)☐ Claim(s) _____ is/are allowed.

6)☒ Claim(s) 1-17 is/are rejected.

7)☐ Claim(s) _____ is/are objected to.

8)☐ Claim(s) _____ are subject to restriction and/or election requirement.

Application Papers

9)☐ The specification is objected to by the Examiner.

10)☐ The drawing(s) filed on _____ is/are: a)☐ accepted or b)☐ objected to by the Examiner.

Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).

Replacement drawing sheet(s) including the correction is required if the drawing(s) is objected to. See 37 CFR 1.121(d).

11)☐ The oath or declaration is objected to by the Examiner. Note the attached Office Action or form PTO-152.

Priority under 35 U.S.C. § 119

12)☐ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).

a)☐ All b)☐ Some * c)☐ None of:

1.☐ Certified copies of the priority documents have been received.

2.☐ Certified copies of the priority documents have been received in Application No. _____.

3.☐ Copies of the certified copies of the priority documents have been received in this National Stage application from the International Bureau (PCT Rule 17.2(a)).

* See the attached detailed Office action for a list of the certified copies not received.

Attachment(s)

1) ☒ Notice of References Cited (PTO-892)

2) ☐ Notice of Draftsperson's Patent Drawing Review (PTO-948)

3) ☒ Information Disclosure Statement(s) (PTO-1449 or PTO/SB/08) Paper No(s)/Mail Date 2.

4) ☐ Interview Summary (PTO-413) Paper No(s)/Mail Date. _____.

5) ☐ Notice of Informal Patent Application (PTO-152)

6) ☐ Other: _____.

DETAILED ACTION

Claim Rejections - 35 USC § 102

1. The following is a quotation of the appropriate paragraphs of 35 U.S.C. 102 that form the basis for the rejections under this section made in this Office action:

> A person shall be entitled to a patent unless –
>
> (e) the invention was described in (1) an application for patent, published under section 122(b), by another filed in the United States before the invention by the applicant for patent or (2) a patent granted on an application for patent by another filed in the United States before the invention by the applicant for patent, except that an international application filed under the treaty defined in section 351(a) shall have the effects for purposes of this subsection of an application filed in the United States only if the international application designated the United States and was published under Article 21(2) of such treaty in the English language.

2. Claims 1, 3, 4, 6, and 13-15 are rejected under 35 U.S.C. 102(e) as being anticipated by Blust et al. (U.S. Patent 6,718,183).

As to claim 1, the Blust reference discloses a method of encoding a communication for transmission over a communications network ("the present invention provides architectures for using vocoders that are designed to improve the quality of the speech that traverses through the architecture" (Col. 2, lines 51-53)), comprising:

selecting one of a plurality of vocoder algorithms ("which format to use depends on many factors, including which vocoder formats are being used, desired speech quality and processing capability of the subscriber units" (Col. 3, lines 29-31)), the selection based on at least one of the following criteria:

a) minimizing bandwidth required to transmit the communication ("the subscriber units decide between themselves which vocoder formats are available to the subscriber units, desired quality, air link bandwidth" (Col. 9, lines 18-21). "In mobile communications, the driver for digital telephony has been increased capacity. To achieve additional capacity in the same channel bandwidth allocations previously used by analog FM systems, it is necessary to use an analog to

digital conversion technique that encodes the speech at a rate much less than 64 kbps" (Col. 1, line 66 to Col. 2, line 4));

b) minimizing a cost of transmitting the communication ("the "high end" of digital telephony is considered to be the conversion of the analog to a digital signals at a rate of 64 kbps" (Col. 1, lines 56-58). "In mobile communications, the driver for digital telephony has been increased capacity. To achieve additional capacity in the same channel bandwidth allocations previously used by analog FM systems, it is necessary to use an analog to digital conversion technique that encodes the speech at a rate much less than 64 kbps" (Col. 1, line 66 to Col. 2, line 4). Reduced rate requires less channel bandwidth and thus minimizes cost of transmission);

c) increasing the quality of the communication ("the subscriber units decide between themselves which vocoder formats are available to the subscriber units, desired quality, air link bandwidth" (Col. 9, lines 18-21). "Another consideration affecting data quality is the particular route the data takes through the intervening network" (Col. 3, lines 44-45). "The present invention can configure the intervening network to minimize impairments to the underlying voice data being transmitted" (Col. 3, lines 52-54));

d) achieving compatibility with a receiving terminal ("the selected vocoder format must be available in each of the elements that the voice data passes through, specifically, the sending subscriber unit, the receiving subscriber unit, and their MSCs" (Col. 3, lines 20-23)); and

e) reducing latency ("another consideration affecting data quality is the particular route the data takes through the intervening network. For example, when using bypass, any non-conforming element in the intervening network will require additional decoding and encoding steps. As described above, these steps degrade the quality of the underlying transmitted voice

signals. Using the common channel signaling provided by SS7, the MSC can assign intervening network elements to handle the call. That is, the present invention can configure the intervening network to minimize impairments to the underlying voice data being transmitted" (Col. 3, lines 44-54). "While certain quality requirements are different for data than voice, i.e., moderate delay is acceptable for data, but corruption of bit order or loss of bits is totally unacceptable" (Col. 2, lines 20-23));

encoding the communication with the selected vocoder algorithm ("subscriber unit 402 has a vocoder 403 that can "impersonate" various vocoder formats 1-N. Subscriber unit 404 has a vocoder 416 that uses vocoder format 2. Vocoder format 2 is one of the vocoder formats subscriber unit 402 can impersonate. Subscriber unit 402 determines that it should use vocoder format 2 to digitize the voice data for transmission to subscriber unit 404" (Col. 7, lines 11-18). See also Figure 4); and

transmitting the communication signal ("the digitized data (in vocoder format 2) is sent to MSC 408 through base station 406" (Col. 7, lines 18-19)).

As to claim 3, the Blust reference discloses the method of claim 1, wherein the communication is transmitted from a calling terminal to a called terminal without converting the communication to a waveform representation ("because the sending and receiving subscriber units use the same vocoder format, no additional decoding/encoding steps need be performed by the vocoders located in MSC 106 and 108. Consequently, vocoders 107 and 109 in MSCs 106 and 108 respectively are bypassed. That is, the voice data from sending subscriber unit 102 is transmitted directly to receiving subscriber unit 102 without being processed by vocoders 105 and 107" (Col. 5, lines 49-56)).

As to claim 4, the Blust reference discloses the method of claim 1, wherein a low bit rate vocoder algorithm is selected if bandwidth is scarce ("as is well-known in the art vocoder algorithms differ in complexity and effective bit rate to achieve varying levels of quality of the voice data as it is subjected to conversions" (Col. 2, lines 44-47). "The subscriber units decide between themselves which vocoder formats are available to the subscriber units, desired quality, air link bandwidth" (Col. 9, lines 18-21). "In mobile communications, the driver for digital telephony has been increased capacity. To achieve additional capacity in the same channel bandwidth allocations previously used by analog FM systems, it is necessary to use an analog to digital conversion technique that encodes the speech at a rate much less than 64 kbps" (Col. 1, line 66 to Col. 2, line 4)).

As to claim 6, the Blust reference discloses the method of claim 1, further comprising: adding error correction to the transmitted communication signal ("data interworking may require rate adaptation, protocol conversion, error correction, etc." (Col. 2, lines 24-25)).

As to claim 13, Figure 4 in Blust shows a smart vocoder (403), comprising:

a memory (403) storing a plurality of vocoder algorithms (Formats 1-N) ("subscriber unit 402 has a vocoder 403 that can "impersonate" various vocoder formats 1-N" (Col. 7, lines 11-13));

a smart vocoder unit (403) selecting an optimal vocoder algorithm from one of the plurality of vocoder algorithms (Formats 1-N) (("which format to use depends on many factors, including which vocoder formats are being used, desired speech quality and processing capability of the subscriber units" (Col. 3, lines 29-31). See also Col. 7, lines 13-18); and

an encoder (403) encoding a communication signal according to the selected vocoder algorithm (see Col. 7, lines 15-19).

As to claim 14, the Blust reference discloses the smart vocoder of claim 13, wherein the smart vocoder unit selects an optimal vocoder algorithm based on at least one of the following criteria:

a) minimizing bandwidth required to transmit the communication ("the subscriber units decide between themselves which vocoder formats are available to the subscriber units, desired quality, air link bandwidth" (Col. 9, lines 18-21). "In mobile communications, the driver for digital telephony has been increased capacity. To achieve additional capacity in the same channel bandwidth allocations previously used by analog FM systems, it is necessary to use an analog to digital conversion technique that encodes the speech at a rate much less than 64 kbps" (Col. 1, line 66 to Col. 2, line 4));

b) minimizing a cost of transmitting the communication ("the "high end" of digital telephony is considered to be the conversion of the analog to a digital signals at a rate of 64 kbps" (Col. 1, lines 56-58). "In mobile communications, the driver for digital telephony has been increased capacity. To achieve additional capacity in the same channel bandwidth allocations previously used by analog FM systems, it is necessary to use an analog to digital conversion technique that encodes the speech at a rate much less than 64 kbps" (Col. 1, line 66 to Col. 2, line 4). Reduced rate requires less channel bandwidth and thus minimizes cost of transmission);

c) increasing the quality of the communication ("the subscriber units decide between themselves which vocoder formats are available to the subscriber units, desired quality, air link bandwidth" (Col. 9, lines 18-21). "Another consideration affecting data quality is the particular

route the data takes through the intervening network" (Col. 3, lines 44-45). "The present invention can configure the intervening network to minimize impairments to the underlying voice data being transmitted" (Col. 3, lines 52-54));

d) achieving compatibility with a receiving terminal ("the selected vocoder format must be available in each of the elements that the voice data passes through, specifically, the sending subscriber unit, the receiving subscriber unit, and their MSCs" (Col. 3, lines 20-23)); and

e) reducing latency ("another consideration affecting data quality is the particular route the data takes through the intervening network. For example, when using bypass, any non-conforming element in the intervening network will require additional decoding and encoding steps. As described above, these steps degrade the quality of the underlying transmitted voice signals. Using the common channel signaling provided by SS7, the MSC can assign intervening network elements to handle the call. That is, the present invention can configure the intervening network to minimize impairments to the underlying voice data being transmitted" (Col. 3, lines 44-54). "While certain quality requirements are different for data than voice, i.e., moderate delay is acceptable for data, but corruption of bit order or loss of bits is totally unacceptable" (Col. 2, lines 20-23)).

As to claim 15, the Blust reference discloses the smart vocoder of claim 14, wherein the smart vocoder unit is located in a communication terminal or a base station ("the subscriber units can be any devices capable of sending data, including for example telephony devices such as, wireline telephone, wireless telephones, personal computers, personal digital assistants ("PDAs"), pagers, etc. The receiving and sending devices can also be switches or base stations on which vocoders required for the present invention are implemented" (Col. 5, lines 10-16)).

Claim Rejections - 35 USC § 103

3. The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all obviousness rejections set forth in this Office action:

> (a) A patent may not be obtained though the invention is not identically disclosed or described as set forth in section 102 of this title, if the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made to a person having ordinary skill in the art to which said subject matter pertains. Patentability shall not be negatived by the manner in which the invention was made.

4. Claim 2 is rejected under 35 U.S.C. 103(a) as being unpatentable over U.S. Patent 6,718,183 to Blust et al. in view of Seo (U.S. Patent 6,452,911).

As to claim 2, the Blust reference discloses the method of claim 1. However, it does not expressly disclose the selection of one of the plurality of vocoder algorithm occurs during a call setup. The Seo reference teaches the selection of one of the plurality of vocoder algorithm occurs during a call setup (see Figure 3 and Col. 3, line 42 to Col. 4, line 26).

Therefore, it would have been obvious to one having ordinary skill in the art at the time the invention was made to modify the method of Blust wherein the selection of one of the plurality of vocoder algorithm occurs during a call setup, as taught by Seo, in order to set up necessary vocoders before a call connection.

5. Claims 7-10 are rejected under 35 U.S.C. 103(a) as being unpatentable over U.S. Patent 6,718,183 to Blust et al. in view of Chen (U.S. Patent 6,215,821).

As to claim 7, the Blust reference discloses the method of claim 6. However, it does not disclose compressing the encoded communication signal before adding error correction. The Chen reference teaches compressing the encoded communication signal before adding error correction (see Col. 5, lines 26-33 and Figure 6).

Therefore, it would have been obvious to one having ordinary skill in the art at the time the invention was made to modify the method of Blust to further comprise compressing the encoded communication signal before adding error correction, as taught by Chen, in order to reduce the communication bandwidth for transmission.

As to claim 8, Blust-Chen discloses the method of claim 7, further comprising: decompressing the encoded communication signal at a called terminal (Chen: see Col. 5, lines 33-42 and Figure 7).

As to claim 9, the Blust reference discloses the method of claim 6. However, it does not disclose the compression is performed by a lossless compressor. The Chen reference teaches the compression is performed by a lossless compressor ("lossless compressor 62" in Col. 5, lines 26-33 and Figure 6).

Therefore, it would have been obvious to one having ordinary skill in the art at the time the invention was made to modify the method of Blust wherein the compression is performed by a lossless compressor, as taught by Chen, in order to have more efficient compression to reduce the communication bandwidth for transmission.

As to claim 10, the Blust reference discloses the method of claim 6. However, it does not disclose selecting one of a plurality of compression algorithms to compress the encoded communication signal. The Chen reference teaches selecting one of a plurality of compression algorithms to compress the encoded communication signal ("it is emphasized that although run-length coding is used here to demonstrate intersource compression in the present embodiment, there are other coding schemes that may provided even more efficient compression than vector run-length coding. For example, a system under the present invention can use Huffman coding or

any other lossless entropy coding techniques to achieve such intersource compression" (Col. 5, lines 11-18)).

Therefore, it would have been obvious to one having ordinary skill in the art at the time the invention was made to modify the method of Blust to further comprise selecting one of a plurality of compression algorithms to compress the encoded communication signal, as taught by Chen, in order to achieve efficient compression to reduce the communication bandwidth for transmission.

6. Claim 11 is rejected under 35 U.S.C. 103(a) as being unpatentable over U.S. Patent 6,718,183 to Blust et al. in view of Nallanathan et al. (U.S. Patent 6,728,295).

As to claim 11, the Blust reference discloses the method of claim 6. However, it does not disclose interleaving the communication signal to defeat jamming. The Nallanathan reference teaches interleaving the communication signal to defeat jamming ("an interleaving encoder encodes the input data symbols into two data series and assigning them to the I-phase and the Q-phase respectively, in a QPSK modulator" (Col. 2, lines 15-18). "The invention facilitates better communication over channels characterized by fading and jamming" (Col. 1, lines 23-25)).

Therefore, it would have been obvious to one having ordinary skill in the art at the time the invention was made to modify the method of Blust to further comprise interleaving the communication signal to defeat jamming, as taught by Nallanathan, in order to facilitate better communication over channels characterized by fading and jamming.

7. Claims 5 and 12 are rejected under 35 U.S.C. 103(a) as being unpatentable over U.S. Patent 6,718,183 to Blust et al. in view of Zinser, Jr. et al. (U.S. Patent Application Publication 2003/0028386 A1).

As to claim 5, the Blust reference discloses the method of claim 1. However, it does not expressly disclose a vocoder algorithm is selected which allows the call to be routed over a low cost network. The Zinser, Jr. reference teaches a vocoder algorithm is selected which allows the call to be routed over a low cost network ("in both commercial and military applications, it is desirable to have conference bridge functionality available when using very low rate (2.4 kb/sec and below) digital communication channels" (page 15, col. 1, paragraph [0244], lines 1-4). "The compressed domain conference bridge is designed to provide most of the services available on a conventional bridge, but maintain full intelligibility for all users (even when there are simultaneous talkers). In addition, multiple types of low-rate vocoder algorithms are supported, including a special hybrid-dual/single talker receiver that will allow a user to hear 2 simultaneous talkers over a single 2400 bit/second channel" (page 15, col. 2, paragraph [0249], lines 4-11). Low rate communication channels functionally correspond to low cost network).

Therefore, it would have been obvious to one having ordinary skill in the art at the time the invention was made to modify the method of Blust wherein a vocoder algorithm is selected which allows the call to be routed over a low cost network, as taught by Zinser, Jr., in order to minimize cost of transmission and still maintain quality and intelligibility of a call.

As to claim 12, the Blust reference discloses the method of claim 1. However, it does not expressly disclose converting the encoded communication signal to a different coding standard by use of a compressed domain transcoder. The Zinser, Jr. reference teaches converting the encoded communication signal to a different coding standard by use of a compressed domain transcoder ("the system and method of the present invention comprises a compressed domain universal transcoder that transcodes a bit stream representing frames of data encoded according

to a first compression standard to a bit stream representing frames of data according to a second compression standard" (Abstract, lines 1-5). See also Figure 2 and page 2, col. 2, paragraph [0033]).

Therefore, it would have been obvious to one having ordinary skill in the art at the time the invention was made to modify the method of Blust to further comprise converting the encoded communication signal to a different coding standard by use of a compressed domain transcoder, as taught by Zinser, Jr., in order to allow mobile units that use different vocoder algorithms to communicate.

8. Claims 16-17 are rejected under 35 U.S.C. 103(a) as being unpatentable over U.S. Patent 6,718,183 to Blust et al. in view of McDonough et al. (U.S. Patent 5,926,786).

As to claim 16, the Blust reference discloses the smart vocoder of claim 14. However, it does not expressly disclose the smart vocoder is incorporated into a DSP. The McDonough reference teaches the smart vocoder is incorporated into a DSP ("the emergence of digital signal processors (DSPs) was an important factor in enabling the real time implementation of vocoder algorithms" (Col. 1, lines 46-48). "A first technique to increase the efficiency in the performance of the vocoding algorithm is a specialized DSP core architecture" (Col. 3, lines 12-14). See also "DSP core 4" in Figure 1).

Therefore, it would have been obvious to one having ordinary skill in the art at the time the invention was made to modify the vocoder of Blust wherein the smart vocoder is incorporated into a DSP, as taught by McDonough, in order to increase the efficiency in the performance of the vocoding algorithms.

As to claim 17, the Blust reference discloses the smart vocoder of claim 14. However, it does not expressly disclose the smart vocoder is included in one or more dedicated ASICs. The McDonough reference teaches the smart vocoder is included in one or more dedicated ASICs ("a method and apparatus for implemented a vocoder in a application specific integrated circuit (ASIC) is described" (Abtract, lines 1-2). "The exemplary embodiment of the present invention described herein is an ASIC implementation of a variable rate CELP algorithm detailed" (Col. 2, lines 57-59). "The vocoding algorithm used for exemplary purposes is the variable rate code excited linear prediction (CELP) algorithm" (Col. 1, lines 53-55)).

Therefore, it would have been obvious to one having ordinary skill in the art at the time the invention was made to modify the vocoder of Blust wherein the smart vocoder is included in one or more dedicated ASICs, as taught by McDonough, in order to increase the efficiency in the performance of the vocoding algorithms.

Conclusion

9. The prior art made of record and not relied upon is considered pertinent to applicant's disclosure.

a. Komaili et al. (U.S. Patent 6,529,730) discloses system and method for adaptive multi-rate (amr) vocoder rate adaptation.

b. Widegren et al. (U.S. Patent 6,374,112) discloses flexible radio access and resource allocation in a universal mobile telephone system.

c. Schorman (U.S. Patent 5,745,854) discloses method and apparatus for dynamically adjusting a maximum number of users on a channel utilizing a voice activity factor.

10. Any inquiry concerning this communication or earlier communications from the examiner should be directed to Duy K Le whose telephone number is 703-305-5660. The examiner can normally be reached on 8:30 am - 5:00 pm.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Edward F Urban can be reached on 703-305-4385. The fax phone number for the organization where this application or proceeding is assigned is 703-872-9306.

Information regarding the status of an application may be obtained from the Patent Application Information Retrieval (PAIR) system. Status information for published applications may be obtained from either Private PAIR or Public PAIR. Status information for unpublished applications is available through Private PAIR only. For more information about the PAIR system, see http://pair-direct.uspto.gov. Should you have questions on access to the Private PAIR system, contact the Electronic Business Center (EBC) at 866-217-9197 (toll-free).

Duy Le
May 10, 2004

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